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10945 7590 05/26/2011 NOKIA CORPORATION c/o Ware, Fressola, Van Der Stuyts & Adolphson LLP Building Five, Bradford Green 755 Main Street, PO Box 224 Monroe, CT 06468			EXAMINER OPSASNICK, MICHAEL N	
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**BEFORE THE BOARD OF PATENT APPEALS  
AND INTERFERENCES**

Application Number: 10/692,290

Filing Date: October 23, 2003

Appellant(s): RAMO ET AL.

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Kenneth Q. Lao  
For Appellant

This is in response to the appeal brief filed February 24<sup>th</sup>, 2011 appealing from the Office action mailed September 8, 2008.

## **EXAMINER'S ANSWER**

### **(1) Real Party in Interest**

The examiner has no comment on the statement, or lack of statement, identifying by name the real party in interest in the brief.

### **(2) Related Appeals and Interferences**

The examiner is not aware of any related appeals, interferences, or judicial proceedings which will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

### **(3) Status of Claims**

The following is a list of claims that are rejected and pending in the application:

Claims pending: 1, 3-41 and 49-56

Claims rejected: 1, 3-41 and 49-56

### **(4) Status of Amendments After Final**

The examiner has no comment on the appellant's statement of the status of amendments after final rejection contained in the brief.

**(5) Summary of Claimed Subject Matter**

The examiner has no comment on the summary of claimed subject matter contained in the brief.

**(6) Grounds of Rejection to be Reviewed on Appeal**

The examiner has no comment on the appellant's statement of the grounds of rejection to be reviewed on appeal. Every ground of rejection set forth in the Office action from which the appeal is taken (as modified by any advisory actions) is being maintained by the examiner except for the grounds of rejection (if any) listed under the subheading "WITHDRAWN REJECTIONS." New grounds of rejection (if any) are provided under the subheading "NEW GROUNDS OF REJECTION."

**WITHDRAWN REJECTIONS**

The following grounds of rejection are not presented for review on appeal because they have been withdrawn by the examiner. The 35 U.S.C. 112 rejection of claims 1,3-48 has been withdrawn. The 102 rejection of claims 15-18 under Sinha et al has been withdrawn.

**(7) Claims Appendix**

The examiner has no comment on the copy of the appealed claims contained in the Appendix to the appellant's brief.

**(8) Evidence Relied Upon**

6,311,154	GERSHO et al	10-2001
7,191,136	SINHA et al	03-2007

**(9) Grounds of Rejection**

The following ground(s) of rejection are applicable to the appealed claims:

Claims 1, 3-14, 19-21,26-37, 39-44,46-48 are rejected under 35 U.S.C. 102 (b) as being anticipated by Gersho et al. (6,311,154).

As to claim 1, Gersho et al. teach segmenting {partitioning or classifying} the audio signal {speech} into a plurality of segments {frames} (partitioning samples of a speech signal into frames, col. 4, lines 25-27) based on the audio characteristics {classes} of the audio signal (classifying the speech signal in each from into one of a plurality of classes, col. 4, lines 25-27); and encoding the segments {frames} with different encoding settings {excitation} (encoding an excitation for the frame using one of a plurality of excitation coding...selected according to the class of the frame, col. 4, lines 30-33).

As to claim 3, Gersho et al. teach characteristics {classes/classifying} include voicing characteristics {voice} in said segments {frames} of the audio signal {speech signal} (classifying the speech signal in each frame into classes, classes include voice frame, col. 4, lines 25-27 & 35).

As to claim 4, Gersho et al. teach characteristics {identifying} include energy

characteristics {presence of energy} in said segments {window} of the audio signal {residual signal} (identifying the location of a window, identifying considers the presence of energy in the residual signal, col. 4, lines 65-67).

As to claim 5, Gersho et al. teaches characteristics {positioning} include pitch characteristics {function of the pitch} in said segments {frames} of the audio signal (positioning the window at a location that is a function of a pitch of the frame, col. 4, lines 59-61).

As to claim 6, Gersho et al. teach segmenting {partitioning} is carried out concurrently {classifying and encoding} to said encoding {coding} (partitioning samples of speech, classifying speech signals into classes, coding a speech signal, col. 4, lines 24-25. The classifying and encoding process may be done concurrently).

As to claim 7, Gersho et al. teach segmenting is carried out before said encoding (partitioning samples of speech, classifying speech signals into classes, coding a speech signal, col. 4, lines 24-25, thus the classifying or segmenting is done before coding).

As to claim 8, Gersho et al. teach plurality of voicing values {voice or unvoiced} are assigned to the voicing characteristics of the audio signal in said segments, and wherein said Segmenting {partitioning} is carried out based on the assigned voicing values (classifying a frame is being one of an unvoiced or voiced, col. 4, lines 52-53).

As to claim 9, Gersho et al. teach a value designated {classifying} to a voiced speech signal and another value designated to an unvoiced signal (classifying a frame is being one of an unvoiced or voiced, col. 4, lines 51-52).

As to claim 10 Gersho et al. teach a value designated {classifier} to a transitional stage between the voice and unvoiced {transitional} signals {frame} (classifier for classifying a transition frame, col. 4, lines 52-55)

As to claim 11, Gersho et al. teach a value designated  $\{(m)=l\}$  to an inactive period {silent frame} in the audio signal {speech} (If  $(m)=l$ , then the current frame is declared a silent frame, col. 15, lines 7-8 & 35-37).

As to claim 12, Gersho et al. teach selecting a quantization mode for said encoding in order to improve the bit allocation and to reduce the parameter update rate, wherein the segmenting is carried out based on the selected quantization mode (col. 3 lines 45-49; Fig. 5 and col. 11 lines 4-16; col. 4, lines 36-37, col. 15, lines 35-36 & col. 9, lines 63-65).

As to claim 13, Gersho et al. teach segmenting is carried out based on target accuracy in reconstruction of the audio signal, wherein the target accuracy is selected based on distortion criteria comparing up-sampled quantized values (transmitted samples) and modified parameters (col. 9, lines 63-65 and col. 3 lines 45-49).

As to claim 14, Gersho et al. teach segmenting is carried out for providing a linear pitch representation in at least some of said segments (col. 9, lines 63-65; col. 3 lines 45-49 and col. 4 lines 50-62).

As to claim 19 and 27, Gersho et al. (154) teach an input for receiving audio data indicative of the parameters in the adjusted representation (input applied to element 14, Fig. 3). and a module responsive to the audio data for generating the audio signal based on the adjusted signals and the characteristics of the audio signal (Fig. 3. One would necessarily need a module to respond to an adjusted audio signal/characteristics of audio signals).

At the time of the invention, it would have been inherent to one of ordinary skill in to use a decoder in order to reverse the encoding data for further processing, such as modulating or storing the audio signal.

As to claim 20 and 28, Gersho et al. (154) teaches recording parameters (col. 29 lines 25-35);

As to claim 21 and 29, Gersho et al. (154) teach.  
the audio data is transmitted through a communication channel and wherein the input of the decoder is operatively connected to the communication channel for receiving the audio data (digital communications, col. 1, line 1 and Fig. 3).



As to claim 26, Gersho et al. (154) teach, a code for determining the characteristics of the audio signal (LP coding, col. 8 lines 54- a code for adjustment the parameter based on the characteristics of the audio signal for providing an adjusted representation of the parameter, wherein said adjusting comprises the steps of segmenting the audio signal into a plurality of segments based on the characteristics of the audio signal and encoding the segments based on one or more of a plurality of encoding settings (LP coding, modified residual, adjusts frames, Abstract and Fig. 9; col. 8 lines 54-63).

As to claim 30, Gersho et al. (154) teaches a mobile terminal (mobile base station, col. 6, lines 17-18).

As to claim 31, Gersho et al. (154) teaches implementing in a cell phone system which necessarily has both base station and mobile station adapted to communicating with the base stations (col. 6, lines 33-36); a decoder for use in parametric audio coding for generating a synthesized audio signal indicative of an audio signal having audio characteristics, wherein the audio signal is coded in a coding step into a plurality of parameters at a data rate and the encoding step is adjusted based on the characteristics of the audio signal for providing an adjusted representation of the parameters, wherein the said adjusting comprises the steps of segmenting the audio signal into a plurality of segments based on the characteristics of the audio signal and encoding the segments based on one or more of a plurality of encoding settings (Figs 1, 4-5, LP coding, modified residual, adjusts frames, Abstract and Fig. 9; col. 8 lines 54-63 ).. an input for receiving audio data indicative of the parameters in the adjusted representation from

at least one of the base stations for providing the audio data to the decoder, so as to allow the decoder to generate the synthesized audio signal based on the adjusted representation (Figs 1, 4-5, col. 3 lines 1-15).

As to claim 32, Gersho et al. (154) teach, an input for receiving audio data indicative of end points defining a plurality of sub-segments, wherein the audio signal is encoded for providing parameters indicative of the audio signal, the parameters including pitch contour data containing a plurality of pitch values representative of an audio segment in time, and wherein the pitch contour data in the audio segment in time is approximated by a plurality of consecutive sub-segments in the audio segment, and wherein the end points include a first end point and a second end point for defining each of said sub-segments (decoder, col. 6 lines 8-11 and Fig. 1); and a reconstruction module for reconstructing the audio segment based on the received audio data (Fig. 9; col. 6 lines 8-11).

As to claim 33, Gersho et al. (154) teach encoding settings inherently include bit allocation (col. 3 lines 45-49), quantization accuracy (Fig. 5 and col. 11 lines 4-16), quantization method (col. 11 lines 4-16) and parameter update rate (col. 3 lines 31-44 and 56-60).

As to claim 34, Gersho et al. (154) teach, the audio signal contains sinusoidal components (col. 3 lines 25-29, analysis windows made equal becomes sine) and said parameters include frequency values (Fig. 1 element 68), amplitude values (col. 3 lines 51-55) and phase

values indicative of the sinusoidal components (Fig. 1 element 76 and col. 3 lines 25-29).

As to claim 35, Gersho et al. (154) teach the parameters includes pitch (col. 4 line 60), voicing  $f$  (Fig. 9 element 42c), amplitude (col. 3 lines 51-55) and energy of the audio signal (col. 3 lines 42-44).

As to claim 36, Gersho et al. (154) teach the parameters include pitch contour data (col. 4 line 60-61) containing a plurality of pitch values inherently representative of an audio segment in time (col. 4 lines 59-63 and col. 2 lines 51-64).

As to claim 37, Gersho et al. (154) teach encoding settings inherently include bit allocation (col. 3 lines 45-49), quantization accuracy (Fig. 5 and col. 11 lines 4-16), quantization method (col. 11 lines 4-16) and parameter update rate (col. 3 lines 31-44 and 56-60, Fig. 4, 8-9 and 14).

As to claim 40, Gersho et al. (154) teach encoding settings inherently include bit allocation (col. 3 lines 45-49), quantization accuracy (Fig. 5 and col. 11 lines 4-16), quantization method (col. 11 lines 4-16) and parameter update rate (col. 3 lines 31-44 and 56-60, col. 6 lines 8-11).

As to claim 41, Gersho et al. (154) teach, wherein the audio signal comprises a plurality of frames and the audio signal in each frame has a waveform and wherein the further audio signal is

produced in the decoding stage independently of the waveform (col. 14 lines 8-14; col. 13 lines 62-67 and col. 14 lines 1-7).

As to claim 42, which depends on claim 1, Gersho et al. (154) teach wherein each segment has a segment length and wherein the segment length of at least one segment is different from the segment length of at least one other segment (col. 14 lines 8-14; col. 13 lines 62-67 and col. 14 lines 1-7).

As to claim 43, which depends on claim 19, Gersho et al. (154) teach wherein the audio signal comprises a plurality of frames and the audio signal in each frame has a waveform and wherein the module generates the further audio signal independently of the waveform (col. 14 lines 8-14, col. 13 lines 62-67 and col. 14 lines 1-7).  
.14).

As to claim 44, which depends on claim 19, Gersho et al. (154) teach wherein the segments comprise segments of different segment lengths (col. 14 lines 8-14).

As to claim 46, which depends on claim 26, Gersho et al. (154) teach wherein the segments comprise segments of different segment lengths (col. 14 lines 8-14).

As to claim 47, which depends on claim 31, Gersho et al. (154) teach

wherein the segments comprise segments of different segment lengths (col. 14 lines 8-14).

As to claim 48, which depends on claim 32, Gersho et al. (154) teach wherein the segments comprise segments of different segment lengths (col. 14 lines 8-14).

Claims 22-25,38,45 are rejected under 35 U.S.C. 102(e) as being anticipated by Sinha et al (7191136).

As per claims 22,23,45, Sinha et al (7191136) teaches a method for use in a parametric audio coding to encode an audio signal by segmenting the audio signal into a plurality of segments based on audio characteristics of the audio signal (by high pass filtering the input audio signal (col. 4 lines 47-51), and then performing a non-linear parametric representation of the signal – col. 4 lines 53-59; wherein the data amount per processing depends upon the frequency characteristics of the audio signal, and the characteristics analyzed can be peak analysis, lattice quantization, or frequency range selection – col. 3 lines 1-6); encoding the segments with different encoding settings (by choosing compression settings on-the-fly → col. 6 lines 43-47); also teaching upsampling (col. 7 lines 42-44) or downsampling (col. 7 lines 39-46).

As per claims 38, Sinha et al (7191136) teaches quantized and unquantized features (col. 3 lines 1-6)

As per claim 24, Sinha et al (7191136) teaches storage mediums (col. 8 lines 1-9).

As per claim 25, Sinha et al (7191136) teaches header information transmitted over communication channels (col. 6 line 64 – col. 7 line 7).

#### **(10) Response to Argument**

As to appellants arguments on pages 12-13, and page 16 –bottom of page 18 (relating to applicants Fig. 4) of the Appeal Brief, examiner notes that the although the 35 U.S.C. 112 Rejection has been withdrawn, the following summarizing point needs emphasis: that the claim scope of claim 1 can have one of the following interpretations:

1) a typical audio encoder that extracts audio signal information (outputting segments based upon voice/unvoiced, silence decision( output of encoder 12, line 112, in Figure 4) from input line 110 into the sub-block 12 in Figure 4, generating segmented audio with associated parameters 112 (detailed as existing speech coder in applicants specification, page 13, lines 8-14; page 14, lines 19-26), (as interpreted by Examiner)

2) a typical encoder generating parameters into compression block 20 of Figure 4, the block 20 re-segments the sequence of initial segments based on the degree of voicing, etc., derived from speech parameters (Figure 4, and Figure 51 page 15, lines 1-17 of the specification), (as interpreted by Appellant).

Examiner argues that the current claim elements of claim 1 do not contain enough linking steps to consider both subblock 12 and subblock 20 (both of Figure 4) with signal path 110 and 112, as the claim scope of claim 1.

- A. "obtaining step" of obtaining parameters from an audio signal.
- B. "partitioning step" of partitioning the audio signal using the parameters
- C. "Encoding step" of encoding the segments

Examiner argues that fig. 4, line 110 into the encoder, and output 112 of parameters are matched to claim 1 as follows – (and seen in applicants spec, page 13, lines 8-13) → the encoder of Fig. 4 performs the encoding step (as outputting parameters), the parameters are a result of analyzing the input audio signal 110, and the "partitioning step" occurs as the typical encoder performs voicing decisions over a measure time period of the input signal (page 5 lines 22-25, and page 11 lines 26-31; both of applicants spec). In addition, applicants spec shows a "pre-processing" stage with measure energy levels and frequency characteristics (page 13, lines 12-13). Examiner argues that **the above matchings represent the claim scope of claim 1 without referring to subblock 20, as argued by appellant.**

Furthermore, appellant argues that the compression block 20 performs "segmentation"; however, the claim 1 states "encoding the segments", however, the output of compression block does not feed-back to encoder 12, nor does compression block 20 show an encoder. Examiner then argues, how can the compression block be considered as the "partitioning into segments" *when there is no re-connection to the encoder for the "encoding the segment step" ?* **Examiner concludes that according to the claim steps of claim 1,**

**compression block 20 cannot be considered as partitioning the audio signal into segments.**

Nonetheless, regardless of which interpretation is chosen, the Gersho reference meets the claim scope of claim 1, as explained below.

As to appellants arguments on pages 13-15, pg 18-19 (Gersho reference) of the Appeal Brief, appellants argue that Gersho performs frame classification only after segmenting or partitioning. Examiner disagrees with Appellants assessment; Examiner points to Gersho, col. 4 lines 24-35; fig. 3. Of essence, is the performance of steps d) and e), wherein the subframe size for the excitation signal is **resized** based upon the previous processing(classifying) of the audio signal. See figure 2, where the “basic subframe”, sized as  $n2-n1$ , is adjusted to either a minimum of  $n2-n1-(d2+d1)$  or a maximum  $n2-n1+(d2+d1)$ ; nonetheless, the subframe **size is readjusted after** the measurement of parameters of the input signal. Or to match claim words, the “time intervals” of claim 1 map to Gersho’s frame, and the “partitioning of segments” map to Gersho's resizing of the subframe. Lastly, in response to applicant's argument that the references fail to show certain features of applicant’s invention, it is noted that the features upon which applicant relies are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

As to appellants arguments on pages 15-16, pgs 19-20 of the Appeal Brief, appellants argue that Sinha does not teach the partitioning into segments; examiner disagrees and argues 1) from the



showing above, applicants claim scope does not pertain to “partitioning into segments” (disqualifying the compression block 20 of applicants Fig. 4); and 2) Sinha teaches the determination of signal characteristics – col. 3, lines 1-6; encoding the segments on the fly with different encoding settings (col. 6 lines 43-47), based upon analyzed parameters (col. 4 lines 47-51; col. 4 lines 53-59). Lastly, in response to applicant’s argument that the references fail to show certain features of applicant’s invention, it is noted that the features upon which applicant relies are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

**(11) Related Proceeding(s) Appendix**

No decision rendered by a court or the Board is identified by the examiner in the Related Appeals and Interferences section of this examiner’s answer.

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,

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